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Abstract

A digital filter is provided for high quality playback of a recorded audio signal. The filter is designed in the frequency domain by establishing a base point and a series of principal points at uniform frequency intervals. The principal points are mirrored by a series of higher frequency mirror points arranged symmetrically about a mid frequency, at half of the sampling frequency of the recorded audio signal. Those principal points within the range of human hearing are given amplitudes that roughly are inversely corresponding to hearing sensitivity at associated frequencies. After the design procedure has been completed, the frequency domain points are mapped into time response coefficients by inverse discrete Fourier transformation. The time response coefficients then are stored for real time convolution with recorded audio samples.